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# PROGRESS REPORT NO. 7

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## INTRODUCTION

This is the seventh progress report on the research and development Task IV. This month's report covers the design of the modulator and demodulator of a pulse duration modulated unit. The circuit was checked on a closed wire basis and tested for modulation linearity, and frequency. Tests were made to determine the minimum detectable deviation, the overall frequency response and the effect of frequency on deviation.

## DISCUSSION

Pulse duration modulation (PDM), is also referred to as pulse length or pulse width modulation. It is a particular form of pulse-time modulation, which has been previously investigated. Pulse duration modulation involves the modulation of a pulse carrier. For this system the pulse carrier frequency was designed to be 7000 cycles per second. Under modulation the duration of each instantaneous sample pulse is varied in proportion to the modulating wave.

The modulating wave may vary the time of occurrence of the leading edge, the trailing edge or both edges of the pulse. Thi unit was designed to have the lagging edge deviate with the si

In pulse duration modulation there is a portion of the pulse that does not vary with signal. Nevertheless this portion is transmitted as signal power. This part of the signal power that carries no information to the receiver is wasted. The amount of power wasted depends upon the maximum extent to which a pulse can be modulated. When the useless part is subtracted from the pulse duration modulated signal, what is left is pulse position modulation, or pulse time modulation. The power saved represents the fundamental advantage of PPM over PDM.

However, in this particular application, the theory is modified to this extent. Pulse time signals require a reference pulse to accompany them, which involves a power waste. Pulse duration signals are complete in themselves and require no reference pulse. The two pulses of the PTM system occupied two microseconds each for a total of four microseconds. The PDM pulse as indicated in Figure 12 occupies 13 microseconds.

Noise can be an important limitation on the usefulness of a PDM system. Normally noise rides above a signal or in the absence of signal appears along the base line. Noise also rides along the leading and lagging edges. If both edges are being used as signals the noise contributions from the two edges will combine. Signal to noise is improved three decibels by employing a single deviating edge rather than having both edges in motion.

By having the edges rise sharply, the extraneous deviation contributed by noise can be dept at a minimum. Since the pulse is of constant amplitude, the method of noise clipping used with the PTM unit is applicable to PDM. Both the top and base of the pulse can be clipped at the receiver and the relatively clean center amplified to the required value.

There are two types of sampling possible with PDM. Uniform Sampling involves a process wherein a uniformly occurring pulse samples a modulating signal which has been converted to the form of pulse amplitude modulation. The second type is Natural Sampling and is the method used in this application. It avoide the need of converting the signal to PAM, as a preliminary step. The time of sampling varies and coincides in time with each trailing edge. The leading edge is fixed and occurs at regular intervals.

The mathematical series that represents duration-modulated pulses with trailing edges modulated by natural samples of a sinusoidal modulating wave is as follows:

$$F_1(t) = K + \frac{M}{2}\cos w_v t + \sum_{m=1}^{\infty} \frac{\sin m w_c t}{m\pi}$$

$$- \sum_{m=1}^{\infty} \frac{J_0(m\pi M)}{m\pi} \sin(m w_c t - 2m\pi K)$$

$$- \sum_{m=1}^{\infty} \sum_{n=\pm 1}^{m=\pm\infty} \frac{J_n(m\pi M)}{m\pi}$$

$$\sin\left(m w_c t + n w_v t - 2m\pi K - \frac{n\pi w_v}{2}\right)$$

$\cos w_v t$ : Modulating wave

$w_c$ : Pulse repetition frequency

K = Ratio, in the absence of modulation, of pulse duration to the interval between pulse centers. This is the duty cycle without modulation.

M = Modulation index

$J_o$, $J_n$ : Bessel functions

In multichannel systems the index of modulation is inherently small due to many channels, and the wanted signal is recovered with negligible distortion. For a single channel system where the modulation index could be substantial, the amount of distortion may be serious. The wanted component is present and harmonics of the modulating wave are absent. The frequencies of the important in-band distortion components are

$$w_c - 2\ w_v,\quad w_c - 3\ w_v,\quad w_c - 4\ w_v,\ \text{etc.}$$

The coefficients of these terms are independent of frequency, but depend upon the magnitude of the modulation index, M.

At low-modulation levels the third order product, $w_c - 2\ w_v$, follows a square law with respect to the modulation index; $w_c - 3\ w_v$ follows a cube law, and so on. The following table illustrates how these products compare with the message wave at three modulation levels. The tabulated values are the decibel ratios of the coefficient of the last term of the series to the coefficient of the second term.

| MODULATION Index M In Percent | PRODUCT $w_c - 2\ w_v$ | $w_c - 3\ w_v$ |
|---|---|---|
| 5 | 28 db | 60 db |
| 10 | 22 | 48 |
| 20 | 16 | 36 |

It is apparent that the modulation index must be small to prevent undue distortion of the message.

An analysis of the spectra of a system using Uniform Sampling indicates that the output contains not only the wanted message wave, but its harmonics as well. With Natural Sampling these harmonics are missing. As with Natural Sampling, the other in-band distortion products of the form $w_c - n\ w_v$ are present. This would lead one to expect a net deterioration of quality when the sampling is regular instead of natural. The available literature indicates this to be true.

## DESIGN OF EQUIPMENT

The following is a description of the equipment designed during the period covered by this report. The pulse duration modulator, Fig. 1, consists of a blocking oscillator that determines the basic pulse repetition rate. This was designed to be 7 kc, as indicated in Fig. 7. This enabled the highest audio frequency that could be sampled to be 3 kc, which is adequate for voice communication.

The second stage is a multivibrator. The input half section normally is non-conducting. The positive pulse from the blocking oscillator triggers the input section and causes the output half section to become non-conducting. As the trigger pulse returns to zero the process is reversed and causes the output section to become non-conducting. The modulating audio signal is superimposed on the trigger pulse. Since the trigger pulse rise time is sharp, the instant of triggering the multivibrator is unaffected by the audio signal. However the trigger pulse lagging edge is relatively slow. As a result, the time at which it reaches the multivibrator cut-off voltage point is varied by the superimposed audio signal. The output of the multivibrator is a pulse duration modulated signal wherein the leading edge is fixed, and the time of occurrence of the lagging edge is varied in accordance with a modulating signal.

The modulating audio signal passes through two audio amplifiers, in cascade. This total audio gain of 30 db. was found to be necessary to accommodate a microphone input. The amplifiers were designed to have a flat frequency response between 100 and 6000 cycles per second. This was required to minimize any effect on the linearity of deviation, which in turn would be a limitation on the overall frequency response. The overall response was designed to be 250 to 3000 cycles per second.

The demodulator chassis, Fig. 2, consists of a series of rejection filters designed to eliminate the pulse repetition frequency. The output is fed to a low pass filter. This filter is designed to cut off at 3000 cycles per second. A low pass filter was chosen as the means of filtering because of the frequency characteristics of a pulse. A fourier analysis indicates that strong harmonics of the pulse repetition frequency are present out to the tenth harmonic. The low pass filter effectively eliminates all the harmonics.

The signal is then passed through a rejection filter which attenuates the pulse repetition frequency residue remaining after the L.P. filter. It is then amplified by an audio amplifier, which feeds it to a pair of series and shunt rejection filters. The final stage is a power amplifier that couples to a loud speaker. The overall frequency response of the demodulator section was designed to be 170 to 2900 cycles per second.

## TESTING

The following tests were made on the individual sections of the equipment. The two cascaded audio amplifiers were found to have a flat frequency response between 100 and 6000 cycles per second.

The result effectively eliminates this section as a possible cause of modulation non-linearity. A plot of pulse duration deviation versus frequency, Fig. 4, indicates that the modulator is inherently free of frequency distortion.

Figure 5 is a plot of audio input versus duration deviation, and audio output versus duration deviation. Both curves indicate a linear relationship. This is a necessary requirement to avoid harmonic distortion. The maximum deviation without distortion is 1.44 microseconds.

Figure 6 is a plot of the overall frequency response of the unit on a closed wire test. It indicates the 6 decibel attenuation points as 200 and 2900 cycles per second.

A test was made to determine the pulse duration deviation that would result in a signal at the speaker with a signal to noise ratio of 2 to 1. This minimum detectable deviation was found to be 8 millimicroseconds. This indicates a dynamic modulation range or maximum to minimum deviation of 175 to 1.

Various oscillograms were taken, and the test points are indicated on Fig. 3 as follows:

Figure 7 - This is the plate of blocking oscillator. The basic pulse repetition interval is indicated as 145 microseconds. This corresponds to a frequency of 7000 cycles per second.

Figure 8 - At the input to the multivibrator, the pulse is shown without a modulating signal. The fast rise time is indicated as 0.1 microseconds. The lagging edge is seen to be 1.2 microseconds. This relatively long decay enables the superimposed audio signal to control the cut-off point of the multivibrator.

Figure 9 - The input to the audio amplifier indicates the maximum signal voltage permissible as 2 volts with the volume control set at maximum. An audio oscillator provided the 1000 cycle signal.

Figure 10 - This indicates the audio with the blocking oscillator pulse superimposed. The audio gain is represented as 30 db.

Figure 11 - The plate of the input section of the multivibrator indicates a negative pulse. This shows that the input section is normally non-conducting.

Figure 2 - The plate of the output section of the multivibrator indicates the duration modulated pulse. Without modulation the pulse width is 13 microseconds.

Figure 13 - The input to the demodulator section, indicates the amplitude of pulse required to properly operate the input stage. For these tests the multivibrator output was divided down a hundred times before it was fed to the demodulator.

Figure 14 - The plate of the pulse amplifier indicates a triangular wave shape. The rectangular pulse is modified by the inductive loading of the low pass filter that follows.

Figure 15 - This is the output of the low pass filter. The 7 kilocycle pulse is still very evident as it rides the 1000 cycle modulating signal.

Figure 16 - The output of the first rejection filter indicates a slight pulse ripple remaining on the signal.

Figure 17 - This is the output of the first audio amplifier. The gain of the stage is shown to be 28 db.

Figure 18 - The output of the series and shunt rejection filters does not indicate any pulse ripple.

Figure 19 - The plate of the power amplifier indicates the voltage gain of this stage as 19 db.

Figure 20 - This is the wave shape across the loud speaker. The curve is a reasonably good reproduction of the input signal.

## CONCLUSIONS AND FUTURE PLANS

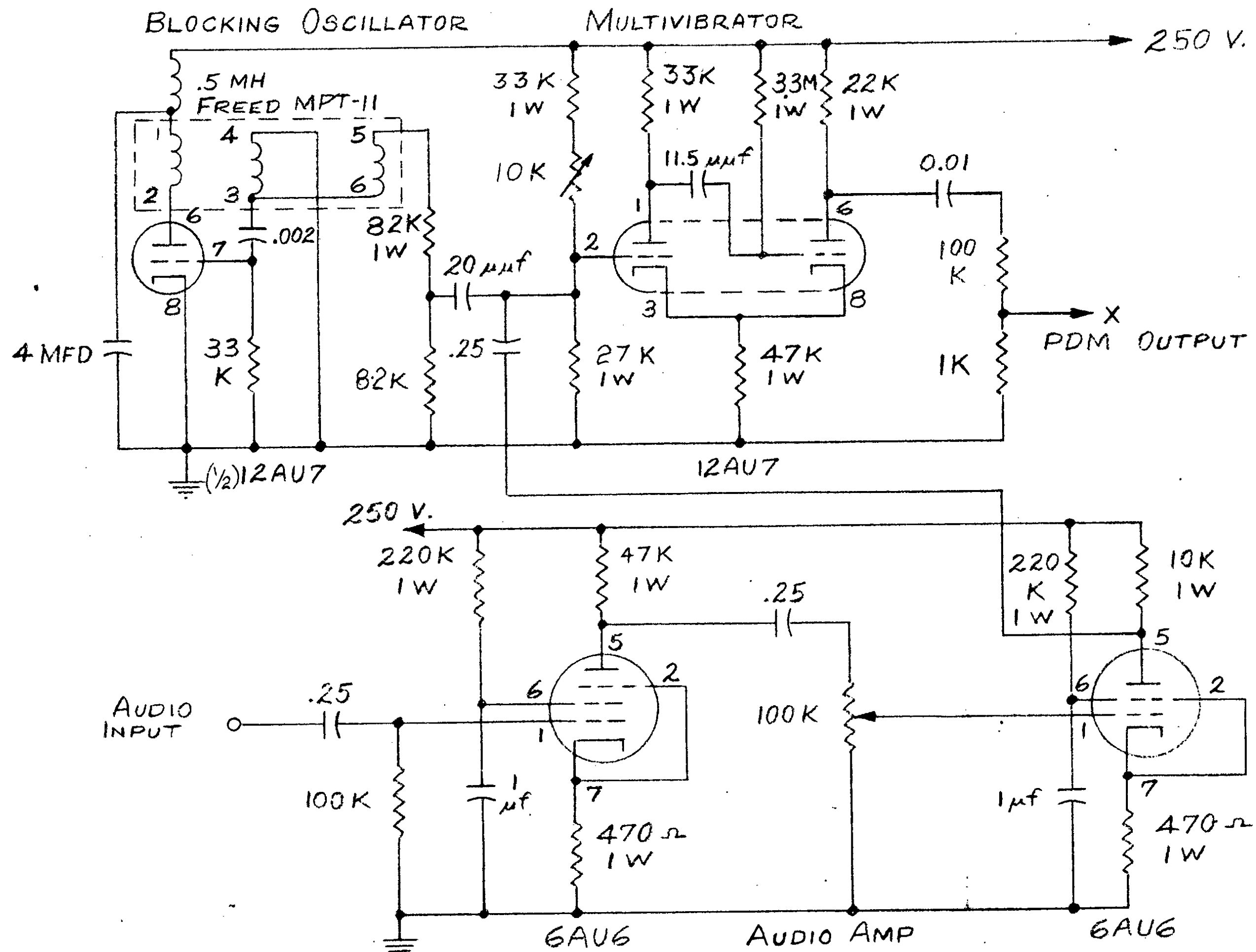
A pulse duration modulation modulator and demodulator was designed and tested.

Tests were made for modulation linearity. It was found that the deviation of the duration modulated pulse varied in a linear manner with the amplitude of the modulating signal. Tests also proved that the deviation was independent of the frequency of the modulating signal within the frequency range used by the unit.

The modulator and demodulator were hooked up as a closed wire system. Tests indicated that the audio output varied in a linear manner with the deviation of the pulse duration. The overall frequency response was 200 to 2900 cycles per second at the 6 db. points. The deviation required to give an audio signal to noise ratio of 2 was found to be 8 milli-microseconds. The maximum deviation for undistorted output was 1.4 microseconds.

It is planned to complete the transmitter and receiver of the pulse duration unit. Tests will be made on a system basis using a radio link.

Tests will then be made to determine the ralative effectiveness of the three systems with regards to communication efficiency and the avoidance of detection on conventional A M - F M receivers.




PULSE DURATION MODULATOR

FIG. 1

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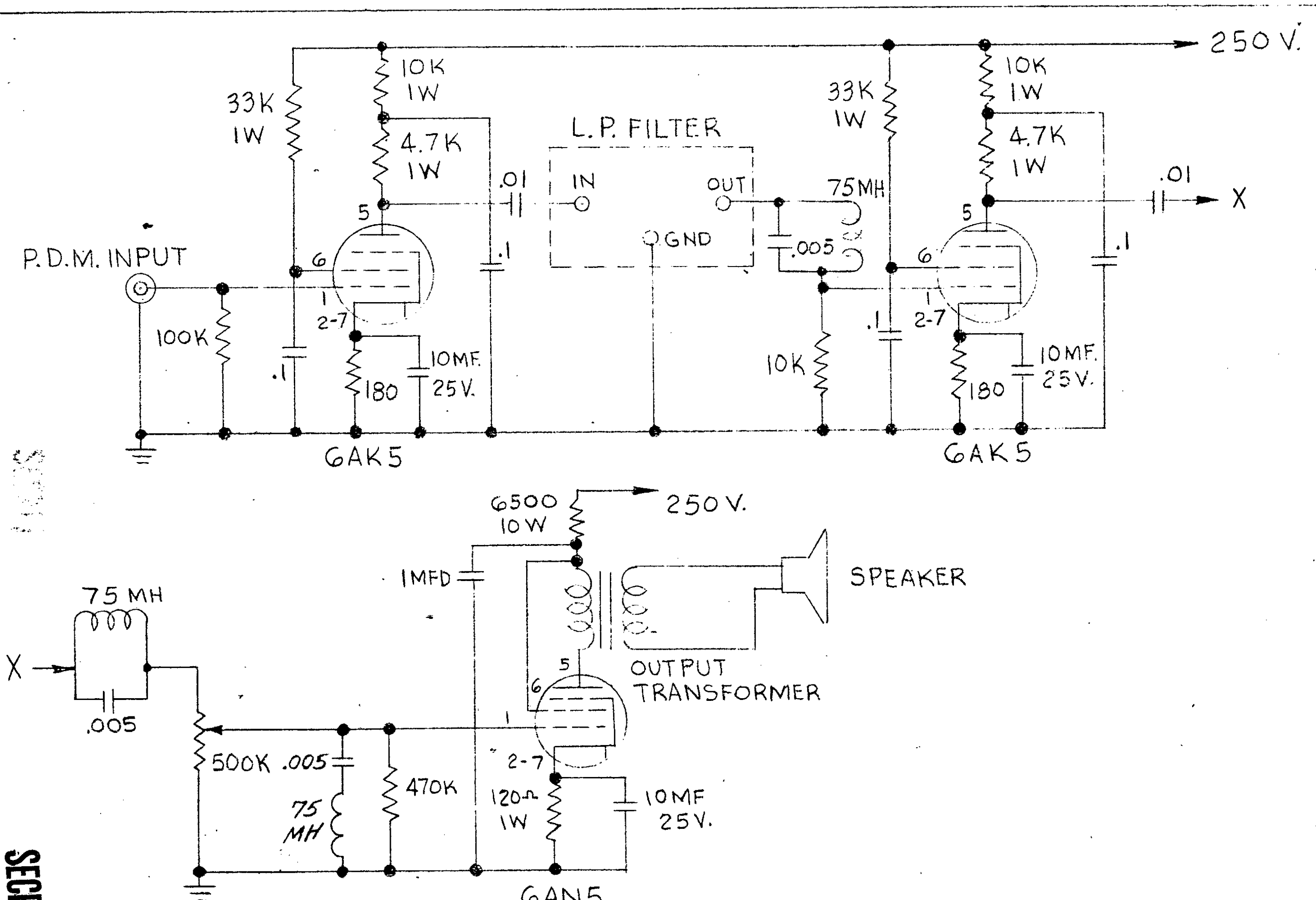


FIG. 2 PULSE DURATION DEMODULATOR

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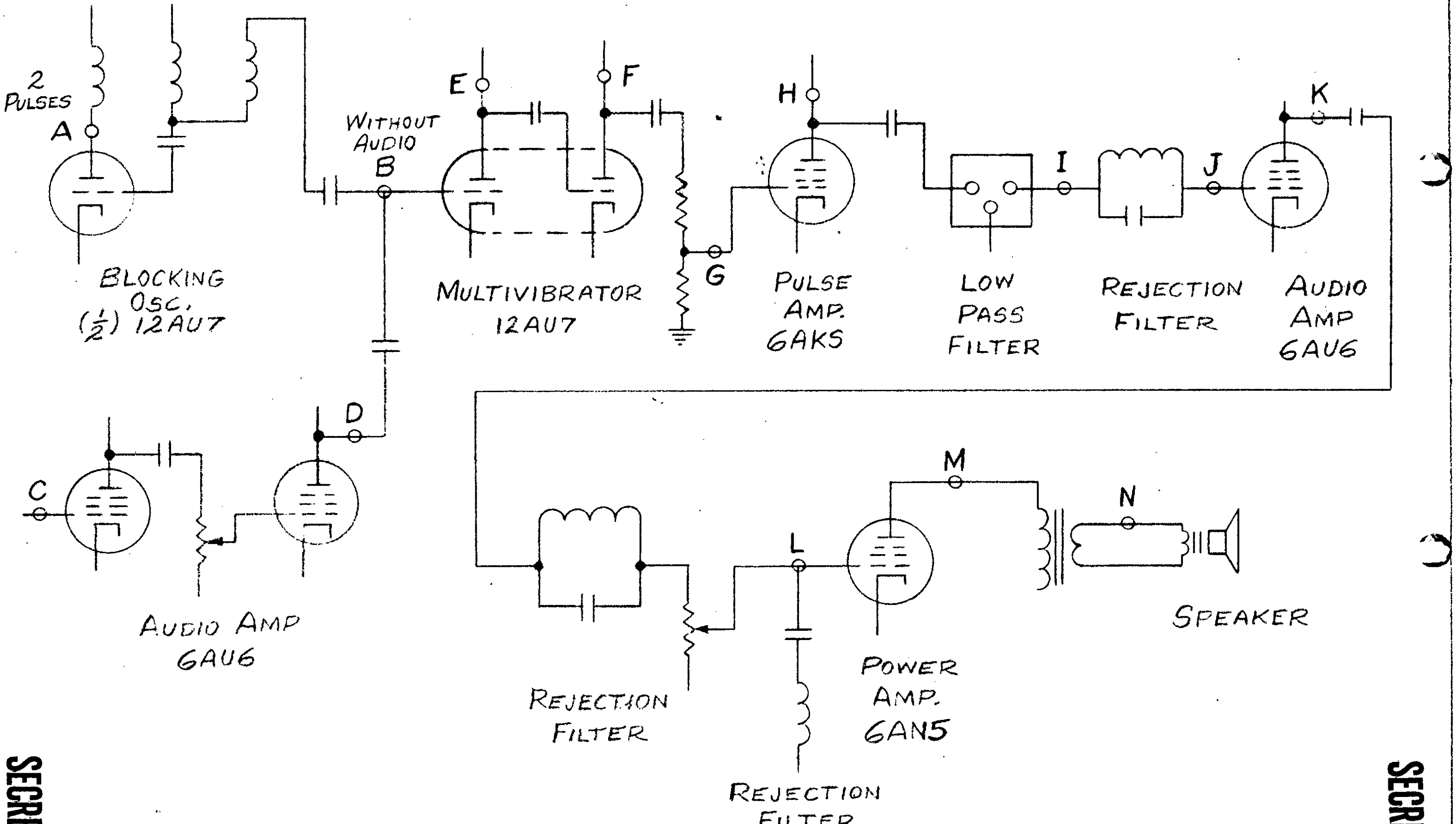


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OUTLINE OF PULSE DURATION MODULATED CLOSED WIRE UNIT

FIG. 3

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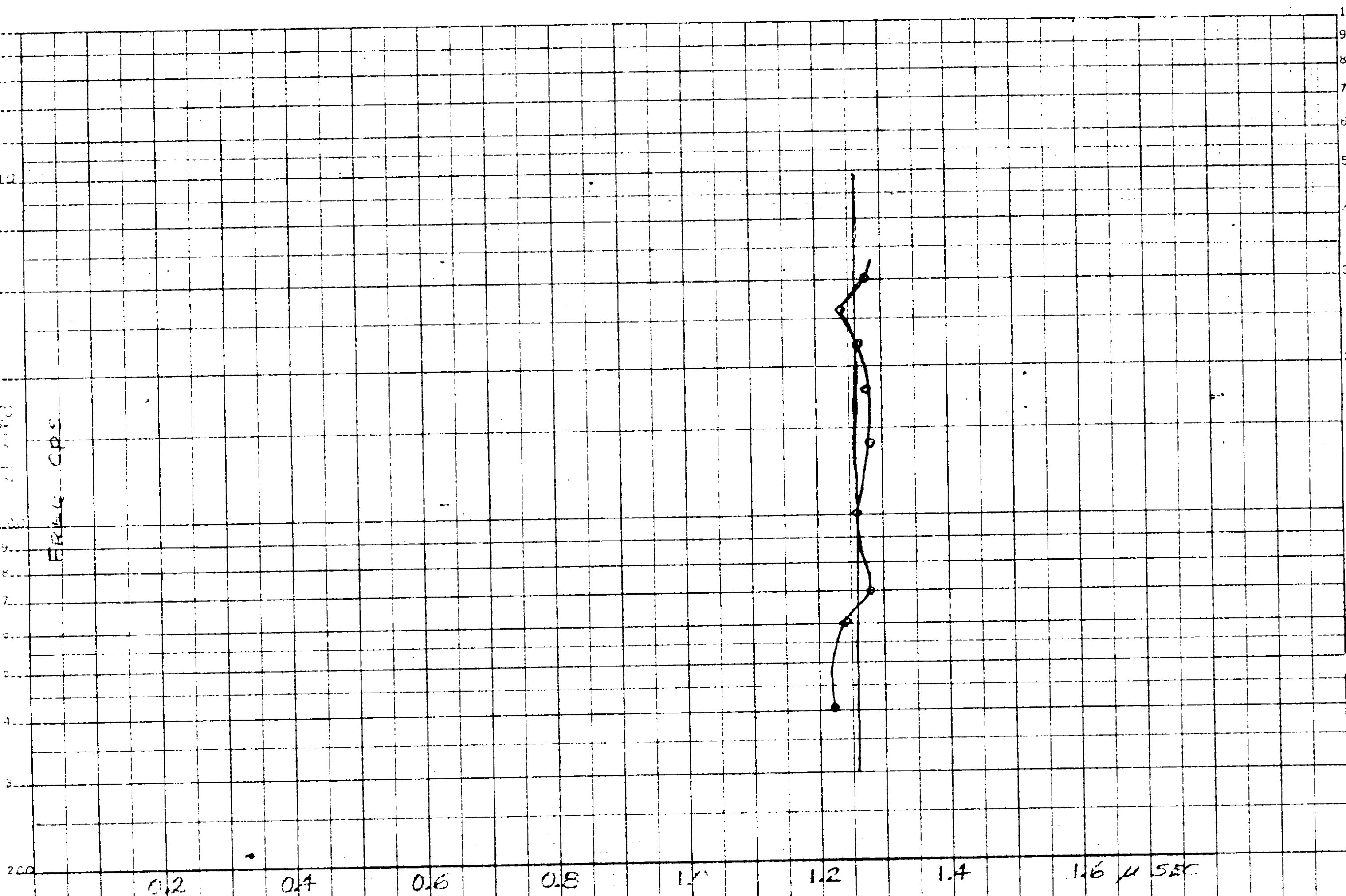


PULSE DURATION MODULATION

PULSE DEVIATION VS FREQUENCY OF AUDIO INPUT SIGNAL

FIG. 4

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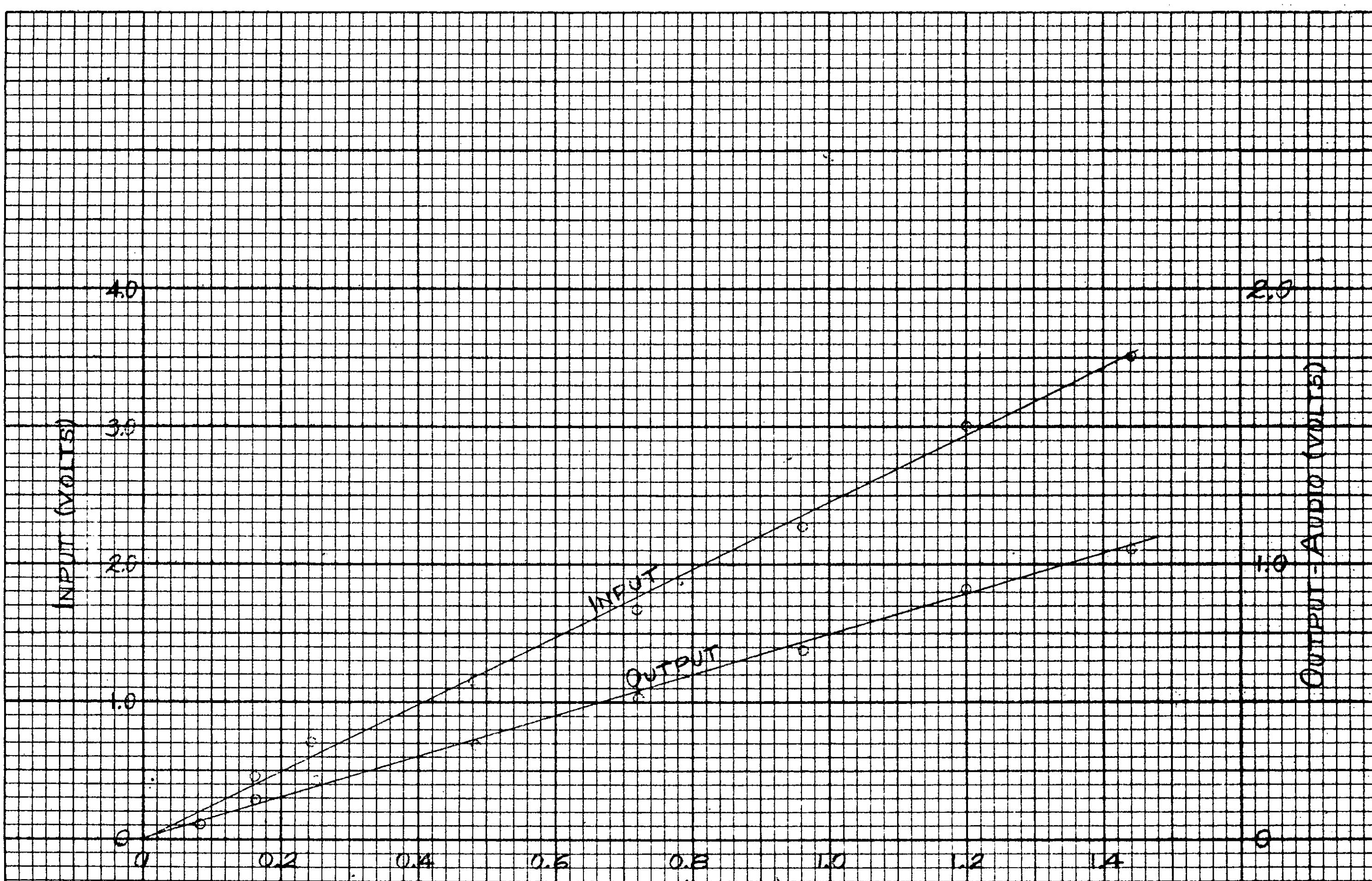
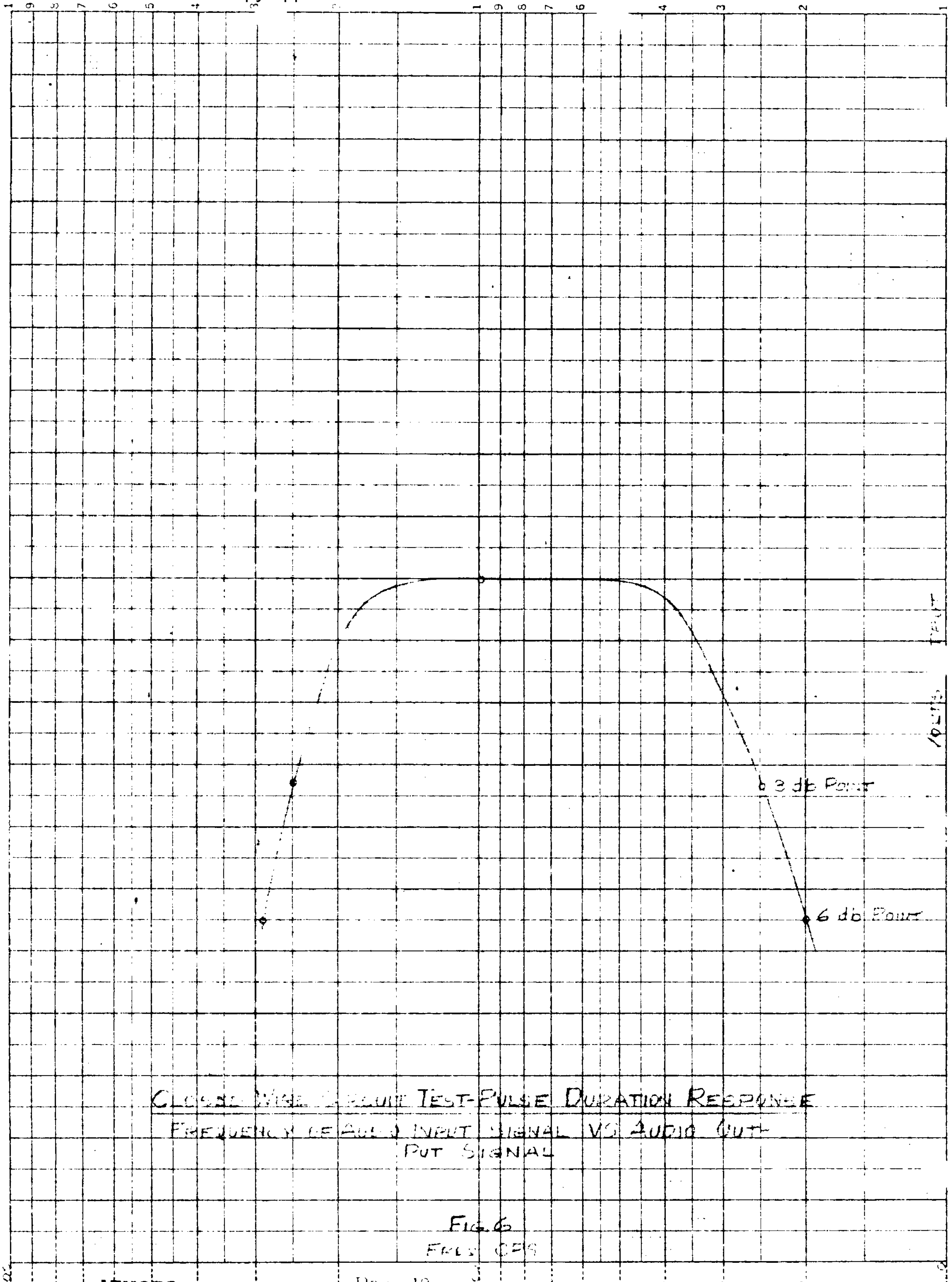



CLOSED WIRE CIRCUIT TEST PULSE DURATION MODULATION
PULSE DEVIATION VS AUDIO INPUT SIGNAL
PULSE DEVIATION VS AUDIO OUTPUT
FIG. 5




Closed Wire Circuit Test-Pulse Duration Response
Frequency of Audio Input Signal VS Audio Out-Put Signal

Fig. 6

Freq CPS



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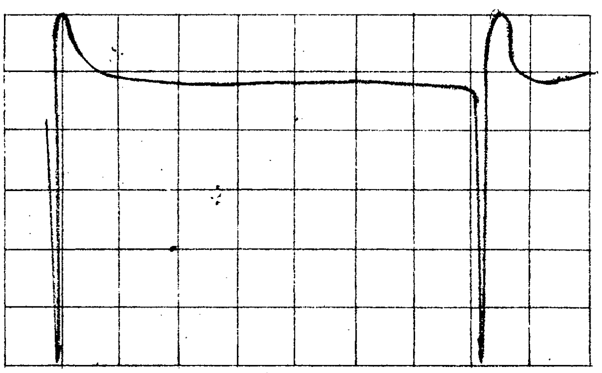

FIG. 7

SENSITIVITY - V/CM 37
SWEEP - $\mu$ SEC/CM 20
SIGNAL A

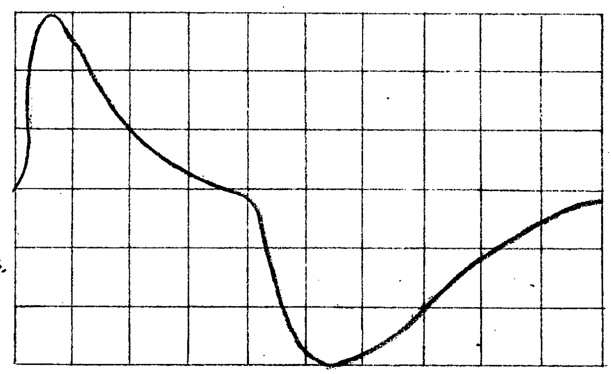

FIG. 8

SENSITIVITY - V/CM 7
SWEEP - $\mu$ SEC/CM 0.5
SIGNAL B (WITHOUT AUDIO)

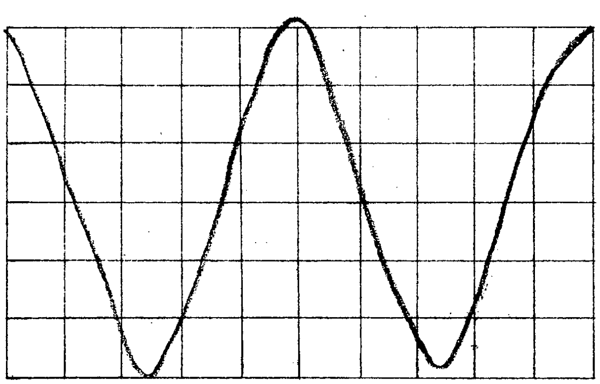

FIG. 9

SENSITIVITY - V/CM 0.3
SWEEP - $\mu$ SEC/CM 200
SIGNAL C

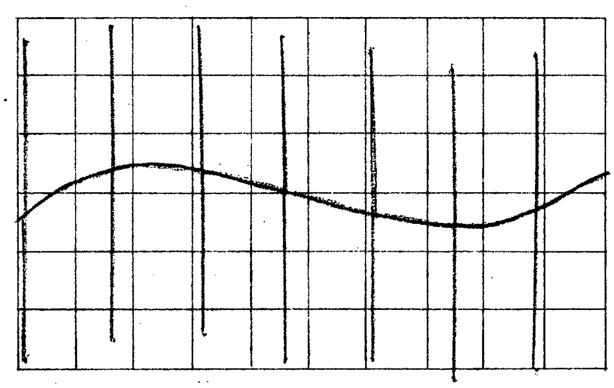

FIG. 10

SENSITIVITY - V/CM 7
SWEEP - $\mu$ SEC/CM 100
SIGNAL D

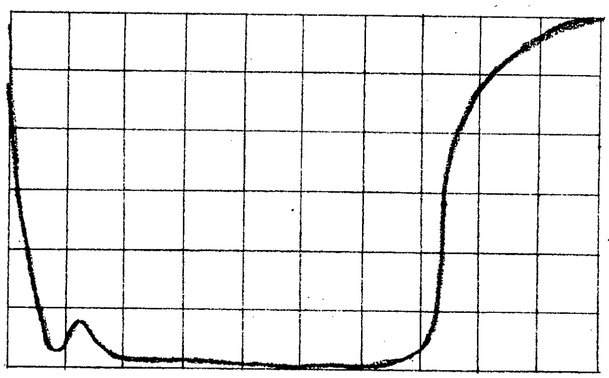

FIG. 11

SENSITIVITY - V/CM 18
SWEEP - μ SEC/CM 1.9
SIGNAL E

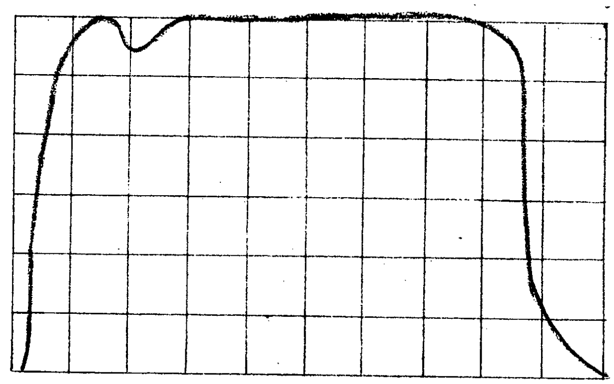

FIG. 12

SENSITIVITY - V/CM 13
SWEEP - μ SEC/CM 1.5
SIGNAL F

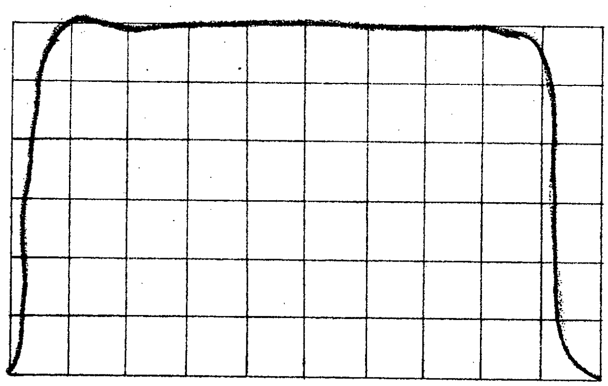

FIG. 13

SENSITIVITY - V/CM 0.13
SWEEP - μ SEC/CM 1.5
SIGNAL G

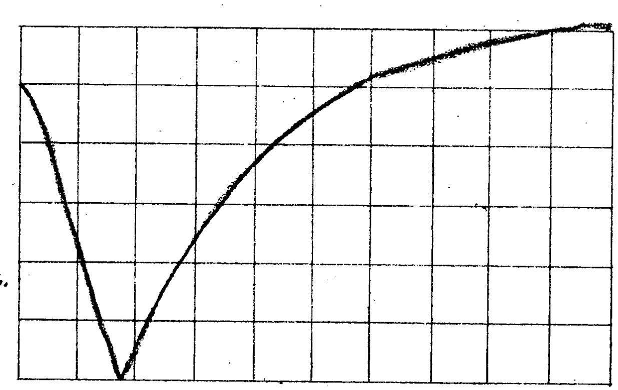

FIG. 14

SENSITIVITY - V/CM 1.3
SWEEP - μ SEC/CM 9.6
SIGNAL H

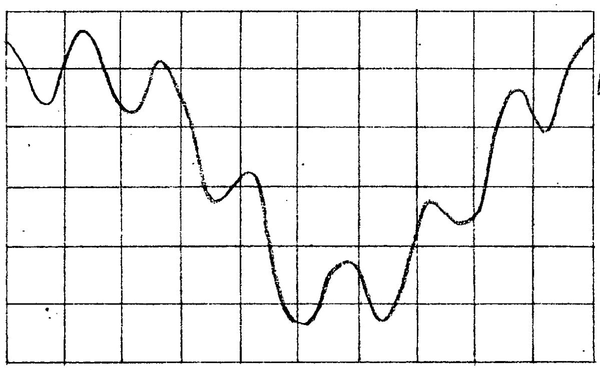

FIG. 15

SENSITIVITY - V/CM 0·03
SWEEP - μ SEC/CM 100
SIGNAL I

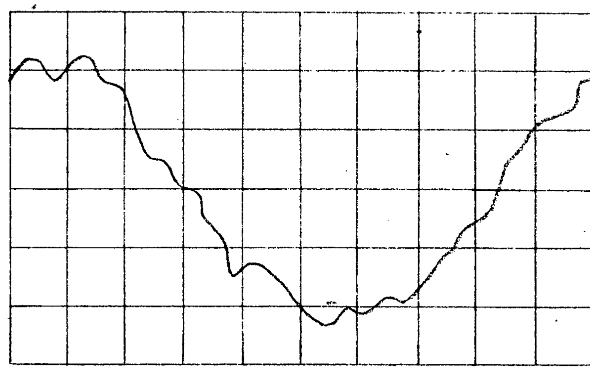

FIG. 16

SENSITIVITY - V/CM 0·03
SWEEP - μ SEC/CM 100
SIGNAL J

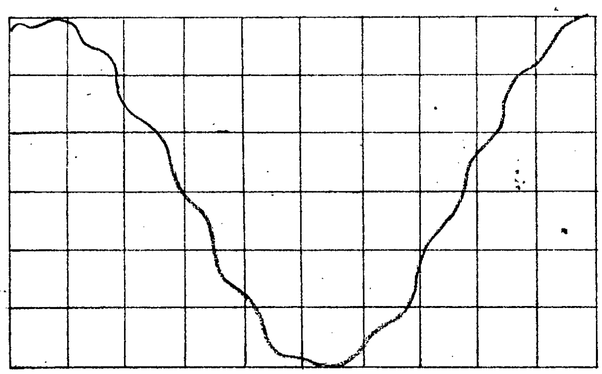

FIG. 17

SENSITIVITY - V/CM 0·7
SWEEP - μ SEC/CM 100
SIGNAL K

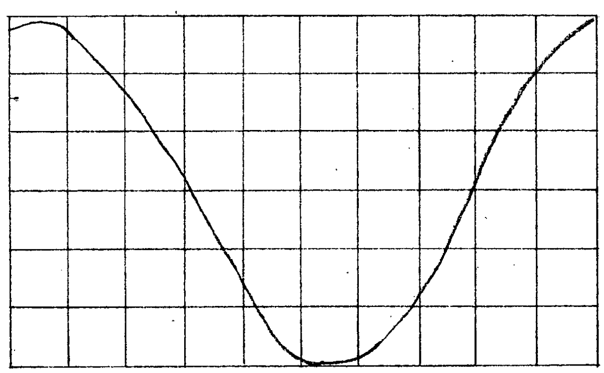

FIG. 18

SENSITIVITY - V/CM 0·6
SWEEP - μ SEC/CM 100
SIGNAL L

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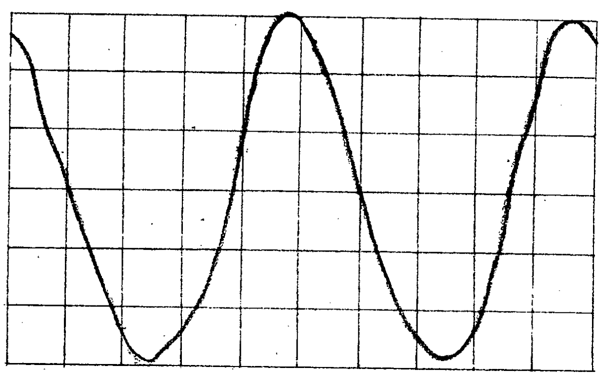

FIG. 19

SENSITIVITY - V/CM 5
SWEEP - μ SEC/CM 200
SIGNAL M

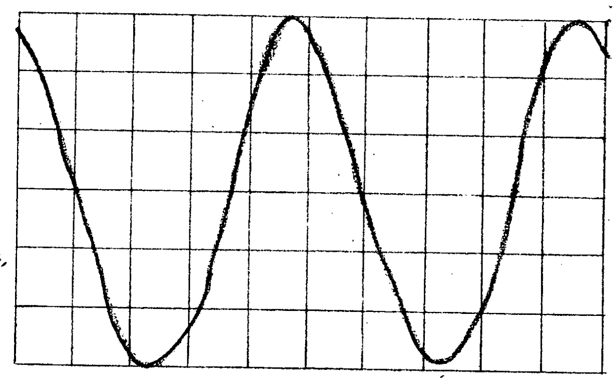

FIG. 20

SENSITIVITY - V/CM 0.5
SWEEP - μ SEC/CM 200
SIGNAL N

SENSITIVITY - V/CM
SWEEP - μ SEC/CM
SIGNAL

SENSITIVITY - V/CM
SWEEP - μ SEC/CM
SIGNAL





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